International Journal of Engineering Research & Technology (IJERT)
ISSN: 2278-0181
Vol. 2 Issue 10, October - 2013

Fading Prediction in Mobile Radio OFDM Systems Using ABICM Method

Pradeep A.V.S [1], Prasad S.N.V.S [2]

[1] *Student (M.Tech), Department of ECE, Chaitanya college of Engineering, Andhra Pradesh, India,*
[2] *Associate Professor, Department of ECE, Chaitanya college of Engineering, Andhra Pradesh, India,*

Abstract

Orthogonal Frequency Division Multiplexing is a scheme used in the area of high-data-rate mobile wireless communications such as cellular phones, satellite communications and digital audio broadcasting. The expression digital communications in its basic form is the mapping of digital information into a waveform called a carrier signal, which is a transmitted electromagnetic pulse or wave at a steady base frequency of alternation on which information can be imposed by increasing signal strength, varying the base frequency, varying the wave phase, or other means. In this instance, orthogonality is an implication of a definite and fixed relationship between all carriers in the collection. Multiplexing is the process of sending multiple signals or streams of information on a carrier at the same time in the form of a single, complex signal and then recovering the separate signals at the receiving end. Adaptive bit-interleaved coded modulation (ABICM) is attractive for rapidly varying mobile radio channels due to its robustness to imperfect channel state information (CSI). A novel ABICM method that exploits the expurgated bound to maintain the target bit error rate (BER) for diverse CSI conditions is proposed and evaluated for an adaptive mobile radio orthogonal frequency division-multiplex (OFDM) system aided by the long-range fading prediction. It is demonstrated that ABICM is much less sensitive to prediction errors than adaptive modulation techniques. Practical channel conditions in spectral efficiency are obtained using ABICM Method.

Index Terms: *Fading channel, OFDM, Interleaved coding, and Adaptive modulation*

1. INTRODUCTION

The OFDM technology was first conceived in the 1960s and 1970s during research into minimizing Inter-Symbol Interference, or ISI, due to multipath. OFDM is a special form of Multi Carrier Modulation (MCM) with densely spaced sub-carriers with overlapping spectra, thus allowing for multiple-access. MCM) is the principle of transmitting data by dividing the stream into several bit streams, each of which has a much lower bit rate, and by using these sub-streams to modulate several carriers. This technique is being investigated as the next generation transmission scheme for mobile wireless communications networks. the advent of the Fourier Transform eliminated the initial complexity of the OFDM scheme where the harmonically related frequencies generated by Fourier and Inverse Fourier transforms are used to implement OFDM systems. The Fourier transform is used in linear systems analysis, antenna studies, optics, random process modeling, probability theory, quantum physics, and boundary-value problems and has been very successfully applied to restoration of astronomical data.

The adaptive bit-interleaved coded modulation (ABICM) was proposed in [1] to improve robustness of adaptive coded modulation to unreliable channel state information (CSI). In the original ABICM method [1], the Bhattacharyya bound based on the minimum distance of the constellation and a nominal non-adaptive BICM scheme were employed to determine the constellation size and the transmission power. The actual BER of this method significantly deviates from the specified target BER [1], [2]. Hence, additional experimental energy adaptation is required to maintain the BER.

ABICM was also investigated in [3], [4] under the assump-tion of perfect CSI at the transmitter, which is reasonable for static fading channels such as indoor wireless systems, but not for outdoor mobile radio channels. Moreover, due to the difficulty of evaluating the exact BER, simulations were used in [3], [4] to obtain the thresholds that determine the transmission constellation as well as power.

The expurgated bound provides accurate BER estimates for non-adaptive BICM in additive white Gaussian noise (AWGN) and fading channels [5]. In this letter we design an ABICM system aided by imperfect CSI by employing the expurgated bound. The ability to maintain the target BER using this method is demonstrated using simulations for an adaptive orthogonal frequency division-multiplex (OFDM) system.

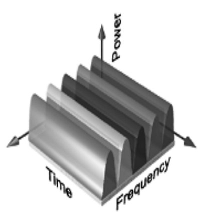
1.1 Multiple Techniques

Multiple access schemes are used to allow many simultaneous users to use the same fixed bandwidth radio spectrum. In any radio system, the bandwidth, which is allocated to it, is always limited. For mobile phone systems the total bandwidth is typically 50 MHz, which is split in half to provide the forward and reverse links of the system.

Sharing of the spectrum is required in order increase the user capacity of any wireless network. FDMA, TDMA and CDMA are the three major methods of sharing the available bandwidth to multiple users in wireless system [2]. There are many extensions, and hybrid techniques for these methods, such as OFDM, and hybrid TDMA and FDMA systems. However, an understanding of the three major methods is required for understanding of any extensions to these methods.

1.1.1 Frequency Division Multiple Access (FDMA):

In Frequency Division Multiple Access (FDMA), the available bandwidth is subdivided into a number of narrower band channels. Each user is allocated a unique frequency band in which to transmit and receive on. During a call, no other user can use the same frequency band.



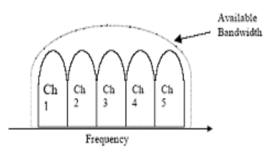


Fig-1 & Fig-2 show the allocation of the available bandwidth into several channels.

1.1.2 Time Division Multiple Access:

Time Division Multiple Access (TDMA) divides the available spectrum into multiple time slots, by giving each user a time slot in which they can transmit or receive. **Fig. 1.4** shows how the time slots are provided to users in a round robin fashion, with each user being allotted one time slot per frame. TDMA systems transmit data in a buffer and burst method, thus the transmission of each channel is non-continuous.

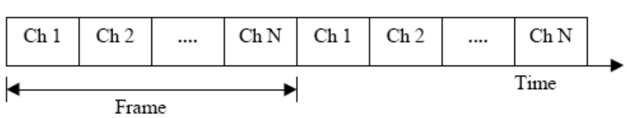


Fig-3 TDMA scheme, where each user is allocated a small time slot

1.1.3 Code Division Multiple Access:

Code Division Multiple Access (CDMA) is a spread spectrum technique that uses neither frequency channels nor time slots. In CDMA, the narrow band message (typically digitized voice data) is multiplied by a large bandwidth signal, which is a pseudo random noise code (PN code) [2]. All users in a CDMA system use the same frequency band and transmit simultaneously. The transmitted signal is recovered by correlating the received signal with the PN code used by the transmitter.

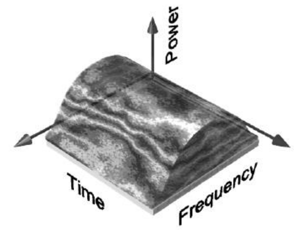


Fig- 4 Code Division Multiple Access (CDMA)

2. OFDM CARRIERS

OFDM is a special form of MCM and the OFDM time domain waveforms are chosen such that mutual orthogonality is ensured even though sub-carrier spectra may over-lap. With respect to OFDM, it can be stated that orthogonality is an implication of a definite and fixed relationship between all carriers in the collection. It means that each carrier is positioned such that it occurs at the zero energy frequency point of all other carriers. The sinc function, illustrated in Fig. 5 exhibits this property and it is used as a carrier in an OFDM system.

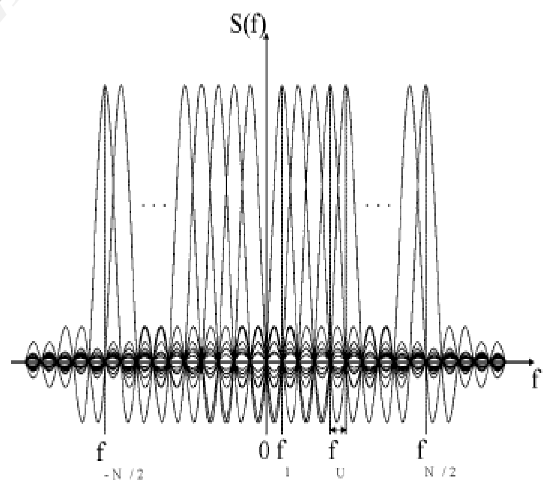


Fig-5 OFDM sub carriers in the frequency domain

2.1 FOURIER TRANSFORM

The *Fourier transform*, in essence, decomposes or separates a waveform or function into sinusoids of different frequencies which sum to the original waveform. It identifies or distinguishes the different frequency sinusoids and their respective amplitudes.

The Fourier transform of f(x) is defined as:

$$F(\omega) = \int_{-\infty}^{\infty} f(x)\, e^{-j\omega x} dx \text{ -------1}$$

and its inverse is denoted by:

$$f(x) = \frac{1}{2\pi} \int_{-\infty}^{\infty} F(\omega) \cdot e^{j\omega x} d\omega \text{ -------2}$$

However, the digital age forced a change upon the traditional form of the Fourier transform to encompass the discrete values that exist is all digital systems. The modified series was called the Discrete Fourier Transform (DFT). The DFT of a discrete-time system, *x(n)* is defined as:

$$X(k) = \sum_{n=0}^{N-1} x(n) \cdot e^{-j\frac{2\pi}{N}kn} \qquad 1 \le k \le N \text{ ----3}$$

and its associated inverse is denoted by:

$$x(n) = \frac{1}{N} \sum_{n=0}^{N-1} X(k) \cdot e^{j\frac{2\pi}{N}kn} \qquad 1 \le n \le N \text{-------}$$

-4

However, in OFDM, another form of the DFT is used, called the Fast Fourier Transform (FFT), which is a DFT algorithm developed in 1965. This "new" transform reduced the number of computations from something on the order of

$$N^2 \text{ to } \frac{N}{2} \cdot \log_2 N. \text{ --------- 5}$$

(a) 2.2 INTER SYMBOL INTERFERENCE

As communication systems evolve, the need for high symbol rates becomes more apparent. However, current multiple access with high symbol rates encounter several multi path problems, which leads to ISI. An echo is a copy of the original signal delayed in time. ISI takes place when echoes on different-length propagation paths result in overlapping received symbols [7]. Problems can occur when one OFDM symbol overlaps with the next one. There is no correlation between two consecutive OFDM symbols and therefore interference from one symbol with the other will result in a disturbed signal

In addition, the symbol rate of communications systems is practically limited by the channel's bandwidth [9]. For the higher symbol rates, the effects of ISI must be dealt with seriously. Several channel equalization techniques can be used to suppress the ISIs caused by the channel. However, to do this, the CIR – channel impulse response, must be estimated.

Recently, OFDM has been used to transmit data over a multi-path channel. Instead of trying to cancel the effects of the channel's ISIs, a set of sub-carriers can be used to transmit information symbols in parallel sub-channels over the channel, where the system's output will be the sum of all the parallel channel's throughputs. This is the basis of how OFDM works. By transmitting in parallel over a set of sub-carriers, the data rate per sub-channel is only a fraction of the data rate of a conventional single carrier system having the same output. Hence, a system can be designed to support high data rates while deferring the need for channel equalizations.

In addition, once the incoming signal is split into the respective transmission sub-carriers, a *guard interval* is added between each symbol. Each symbol consists of a useful symbol duration, T_s and a guard interval, Δt, in which, part of the time, a signal of T_s is cyclically repeated. This is shown in Fig. 2.

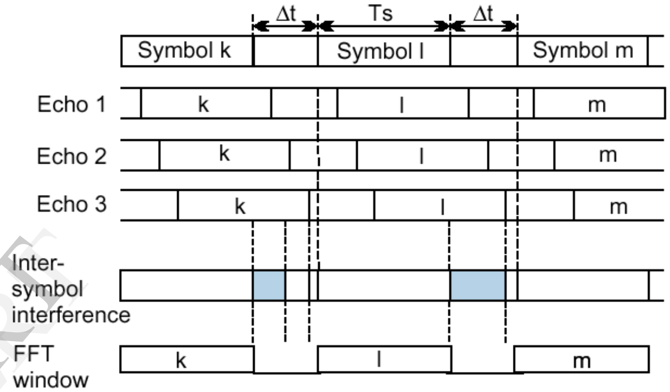


Fig.-6 Combating ISI using a guard interval

As long as the multi path propagation delays do not exceed the duration of the interval, no inter-symbol interference occurs and no channel equalization is required.

3. ADAPTIVE OFDM AIDED BY LONG-RANGE PREDICTION

3.1 Channel Model and Long-Range Prediction

The system diagram of the adaptive OFDM system under investigation is shown in Fig-.7 We assume the frequency selective wide-sense stationary (WSS) Rayleigh fading channel as in [1]. The received signal for the *lth* subcarrier ($l = 1. . . L$) Of the *nth* OFDM symbol is

Y (n, l) = H(n, l)X(n, l) +W(n, l)----- 6

where $H(n, l)$, $X(n, l)$, and$W(n, l)$ are the complex Gaussian channel response $H(n, l) \sim CN(0, 1)$, the transmitted signal, and the complex additive white Gaussian noise with variance $N0$, respectively. It is assumed the inter-symbol interference

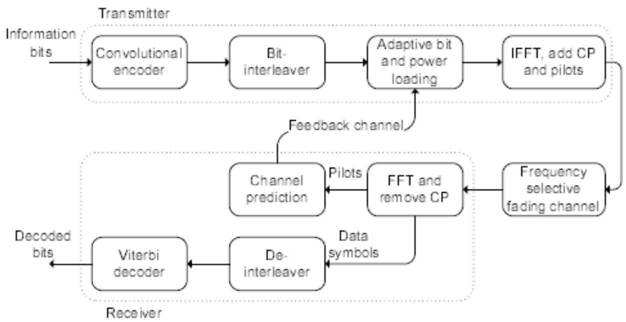


Fig-7 System diagram of LRP-enabled adaptive OFDM system with ABICM.

(ISI) is removed using an appropriate cyclic prefix. Without loss of generality, we set $X(n, l) = E_p$ for pilot symbols, where E_p is the pilot symbol energy.

To facilitate the LRP, pilot symbols are inserted in both frequency and time domains. Past pilot observations within a rectangular area that includes $(2P_f + 1)$ pilot tones and P_t past pilot OFDM symbols are employed to predict the current channel coefficient $H(n, l)$ [2, Fig. 4.2]. In this letter, we assume that channel statistics are known and construct a linear minimum mean square error (MMSE) predictor of order $(2P_f +1)P_t$ [2]. In practice, similar prediction accuracy can be achieved by the auto-regressive (AR) model based predictors that track fading channel variations. The performance of this predictor can be improved at low and medium SNR by employing noise reduction [2], [9], [10]. However, we do not utilize noise reduction since we focus on robustness of adaptive coding methods to imperfect predictions. The channel coefficient $H(n, l)$ and its linear MMSE prediction.

3.2 Adaptive Bit and Power Loading

At the transmitter, the information bits are encoded using a fixed-rate convolutional encoder, followed by bit-interleaving (Fig. 7). To facilitate analysis, we assume that this interleaved is ideal as in [1], [5]. Due to frequency diversity in OFDM systems, this assumption is realistic even for short interleaving depth in time.

After interleaving, the adaptive bit and power loading algorithm maps the coded bits into M-quadrature amplitude modulation (MQAM) symbols for all subcarriers. The constellation sizes and energies of these symbols are determined using the CSI fed back from the receiver. Without loss of generality, we consider the allocation for one OFDM symbol $X(n, l)$. Given predicted channel coefficient ˆH (n, l), $l \in [1, L]$, suppose the minimum average symbol energy required to transmit ml coded bits/symbol while maintaining the target BER is E ˆH $(n,l)(ml)$. Our objective is to maximize the spectral efficiency under the overall energy constraint E_T , i.e.,

$$\max\left\{\sum_{l=1}^{L} m_l = \sum_{l=1}^{L} E_{H(n,l)}(m_l) \le E_T\right. \text{-----}6$$

if $E_{H(n,l)}$ is known as discrete waterfilling algorithm based on the BER calculation for the values of 0.001 and 0.1 using Adaptive Trellis coded modulation (ATCM). The ATCM technique explains about the spectral efficiency with Signal-to-Noise ratio (SNR) with different adaptive values. the MMSE values are obtained with Gaussian distribution values using 0.001 and 0.1. The ATCM obtains the highest spectral efficiency for medium to high SNR. The spectral efficiency using Adaptive bit-interleaved coded modulation (ABICM) is obtained using prediction range [11]. The prediction of data transmission for Adaptive trellis coded modulation (ATCM) for long range prediction (LRP) is obtained in terms of spectral efficiency and Normalized prediction range [8]. The different algorithms of ATCM and ABICM are used with different Gaussian distribution values for obtaining spectral efficiency values. the prediction factor of “K” values are taken for obtaining the spectral efficiency. Comparison of the ATCM and ABICM which obtains the prediction in range for different long prediction range (LRP).

4. RESULTS

The plot explains about the spectral efficiency of ATCM with Gaussian distribution factor 0.001 and 0.1 which is obtained with SNR variations.

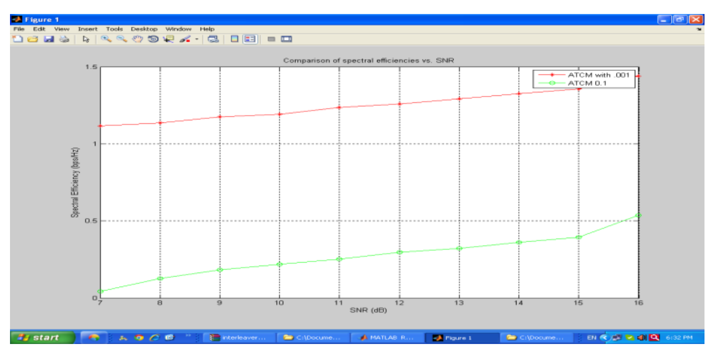


Fig-8 Spectral efficiency for ATCM with 0.001 and 0.1

The plot explains about the spectral efficiency of Adaptive bit-Interleaved coded modulation (ABICM) in long prediction range (LRP) with normalized prediction range.

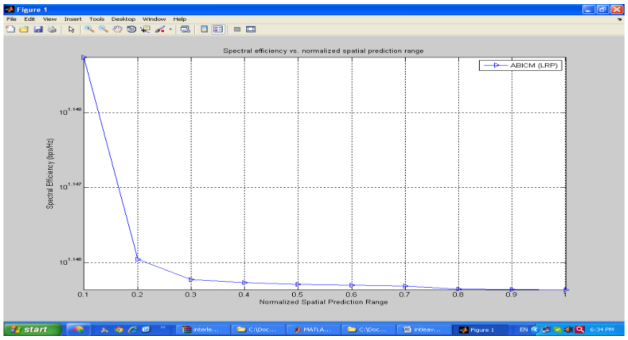
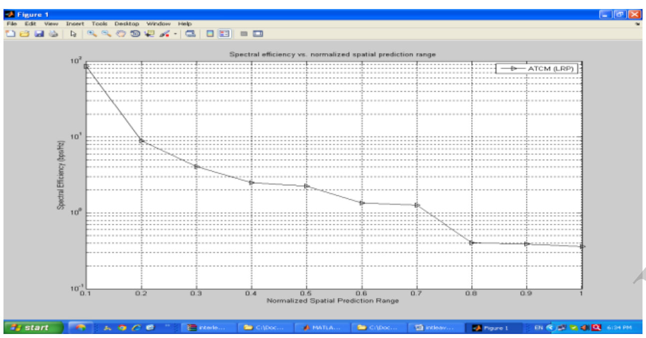


Fig-9 Spectral efficiency of ABICM with LRP

The plot explains about the spectral efficiency of Adaptive trellis coded modulation (ATCM) in long prediction range



(LRP).

Fig-10 spectral efficiency of ATCM of LRP

The plot explains the spectral efficiency with Adaptive bit-interleaved coded modulation (ABICM) with factor k=2 and 4 with Gaussian distribution values of 0.1 and 0.01.

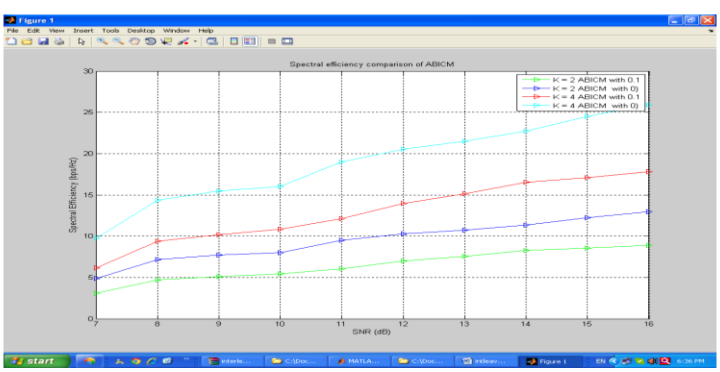


Fig-11 Spectral efficiency of ABICM with different K values

The plot explains about the BER calculation of using different modulation techniques of QAM and QPSK.

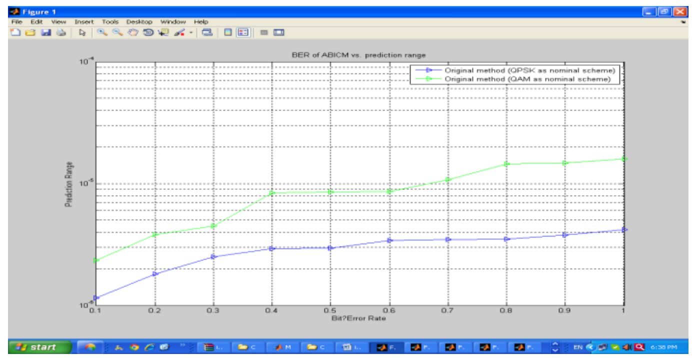


Fig-12 BER calculation using different techniques

5. CONCLUSION

This paper explains about the spectral efficiency calculation using different algorithms of Adaptive trellis coded modulation (ATCM) and Adaptive bit-Interleaved coded modulation (ABICM) using different constant factor values of Gaussian distribution values. The BER calculation explains about the prediction range for different modulation techniques of QAM and QPSK. We can extend the paper for the soft computing techniques with different modulation implementation in future.

REFRENCES

[1] P. Ormeci, X. Liu, D. L. Goeckel, and R. D. Wesel, "Adaptive bitinterleaved coded modulation," *IEEE Trans. Commun.*, vol. 49, no. 9, pp. 1572–1581, Sep. 2001.

[2] T. Jia, "Single and multicarrier adaptive transmission systems with longrange prediction aided by noise reduction," Ph.D. dissertation, NC State University, Raleigh, NC, 2008.

[3] V. K. N. Lau, "Performance analysis of variable rate: symbol-by-symbol adaptive bit interleaved coded modulation for Rayleigh fading channels," *IEEE Trans. Veh. Technol.*, vol. 51, no. 3, pp. 537–550, May 2002.

[4] K. Song, A. Ekbal, S. T. Chung, and J. M. Cioffi, "Adaptive modulation and coding (AMC) for bit-interleaved coded OFDM (BIC-OFDM)," *IEEE Trans. Wireless Commun.*, vol. 5, pp. 1685–1694, July 2006.

[5] G. Caire, G. Taricco, and E. Biglieri, "Bit-interleaved coded modulation,"
IEEE Trans. Inf. Theory, vol. 44, no. 3, pp. 927–946, May 1998.
[6] A. Duel-Hallen, S. Hu, and H. Hallen, "Long-range prediction of fading
signals," *IEEE Signal Process. Mag.*, vol. 17, no. 3, pp. 62–75, May 2000.
[7] A. Duel-Hallen, "Fading channel prediction for mobile radio adaptive transmission systems," *Proc. IEEE*, vol. 95, pp. 2299–2313, Dec. 2007.
[8] A. J. Goldsmith and S.-G. Chua, "Adaptive coded modulation for fading
channels," *IEEE Trans. Commun.*, vol. 46, no. 5, pp. 595–602, May 1998.
[9] T. Jia, A. Duel-Hallen, and H. Hallen, "Improved long-range prediction with data-aided noise reduction for adaptive modulation systems," in *Proc. 2008 Conference on Information Sciences and Systems*.
[10] ——, "Performance of adaptive coded modulation enabled by longrange
fading prediction with data-aided noise reduction," in *Proc. 2008 IEEE MILCOM*.
[11] J. Campello, "Optimal discrete bit loading for multicarrier modulation systems," in *Proc. 1998 IEEE ISIT*, p. 193.

BIOGRAPHIES

A.V.S pradeep pursuing M.Tech in specialization of Digital Electronics and communication systems at Chaitanya Engineering college in Department of Electronics and Communication Engineering.

S.N.V.S. Prasad completed Ph.D and working as an Associate Professor in Chaitanya Engineering college in Department of Electronics and communication Engineering